An Adaptive Lineariser for High Power Amplifiers

Synopsis

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ABSTRACT

The high power amplifiers (HPAs) for CDMA and OFDM telecommunication systems need to be highly linear. However in practice, the response of HPA is non-linear which introduces distortion in signals being amplified. This response can be made linear using various linearization techniques. In this thesis, an adaptive pre-distortion based linearization technique is introduced. It is based on adding pre-distortions into signals, which will result in cancellation of non-linear distortions appearing in power amplifier. The characteristics of this non-linear adaptive lineariser are represented by an analogue polynomial. In this thesis, the coefficients of this polynomial are adjusted automatically in a manner that the output of lineariser when fed to an amplifier, the final output becomes error free and the whole system behaves like a linear amplifier. This helps reducing the distortion in power amplifiers, resulting in a better linearised output. In particularly, a perturbation based approximation method is used to adjust the coefficients. Perturbation based approximation method for adaptive lineariser is based on changing the coefficients step by step and checking the error; whether it is decreasing or increasing. Depending upon the amount of distortion the algorithm may take a few to a large number of cycles before the final output of the overall system (amplifier and lineariser) becomes distortion free. The advantage of this system over the existing (digital) ones is that it has a wider frequency range and lower power consumption. The disadvantage is that the system is slow in nature. In this research, an adaptive lineariser is made using matlab and the results are taken for different values of non-linearities and the optimum value of step size is calculated.

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LIST OF ABBREVIATIONS

CDMA	Code Division Multiple Access
CNR	Carrier to Noise Ratio
EVM	Error Vector Magnitude
FDMA	Frequency Division Multiple Access
HPA	High Power Amplifier
IMD	Inter modulation Distortion
OFDM	Orthogonal Frequency Division Multiple
PA	Power Amplifier
PD	Pre-distortion
SR	Spectral Regrowth
WCDMA	Wideband Code Division Multiple Access
QPSK	Quadrature Phase Shift Keying

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CHAPTER 1

INTRODUCTION

Over the last few years, modern communication services such as OFDM, have created a demand for highly linear high power amplifiers (HPAs). It is primarily, because with the evident of modern bandwidth efficient modulation (BEM) schemes, various services such as video and data can now be transmitted in the same frequency space. It is very important in practice, as these services involve the transmission of multiple signals and/or large quantities of information at high data rates. For such signals, whether transmitted by frequency division multiple access (FDMA), code division multiple access (CDMA) or time division multiple access (TDMA), amplifier linearity is a major consideration. This is explained below.

1.1 Need for linearity

In HPAs, distortion of signals is usually produced by its non-linear response. This distortion is usually measured in terms of amplitude, although it also includes associated abnormalities in phase. For phase, the associated distortion is called phase non-linearity. The shift in phase angle that a signal encounters in passing through an amplifier is a measure of the time delay. For simplicity, phase distortion is not taken into consideration in this synopsis. The term *amplitude linearity*, mentioned above, can be considered a measure of how closely the input-output transfer response of an amplifier resembles a straight-line. However in practise, this response of HPA is usually non-linear and this deviation from the desired straight-line behavior can be represented by the following power series:

$$V_{out} = K_1 V_{in} + K_2 V_{in}^2 + K_3 V_{in}^3 .... + K_n V_{in}^n \qquad (1)$$

Where, $K_1, K_2, K_3 ... K_n$ are the coefficients of the non-linear polynomial, and V_{in} and V_{out} are the respective input and output of HPA. In situation, when a single carrier input signal, (say, a sine wave), is substituted into this expression (Eqn. 1) the output waveform will contain the original sine wave and its harmonic distortion products. The harmonics can be eliminated by filtering and do not pose a problem except for wideband communication

applications requiring wide bandwidth. However, when more than one carrier is present, additional new signals known as intermodulation distortions (IMD) are produced in the vicinity of the input signals. The IMDs are located at frequencies above and below the input carriers as shown in Figure 1.1, and at frequency intervals equal to the separations of the input carriers. Filtering cannot easily eliminate IMD products, as these are located on the same frequency or near to the desired input signals.

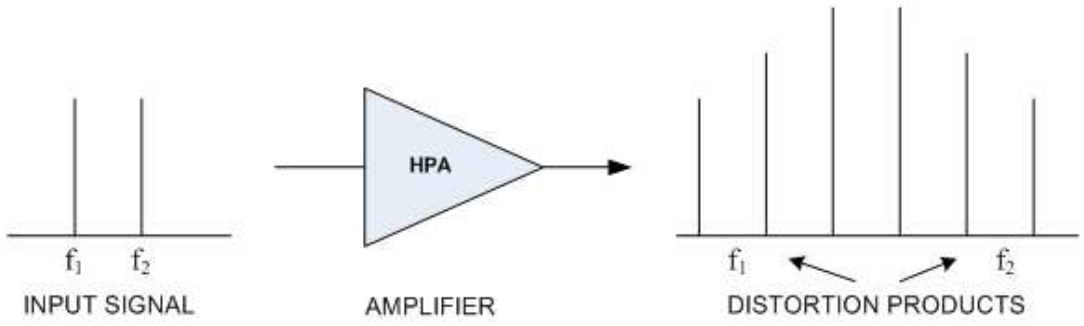


Figure 1.1: A typical example showing that when two signals are amplified using HPAs, the distortion products appear in the vicinity of input signals [1].

From Figure 1.1, it is evident that non-linearity within the system components of HPAs cause distortion of the amplified signals and results in the generation of signals outside of the intended frequency channel or band. These unwanted distortion products are potential interfering sources to further connecting systems and must be reduced to a level, where all the further connecting systems can operate satisfactorily [1,2].

In case of the high power broadcast transmitters, this requirement becomes acute, as the distortion products, although may be many times smaller than the main output signal, may still be quite large in absolute terms and hence cause interference [3]. These distortions frequently become problematic at high drive levels. As an amplifier approaches compression, it imparts both phase and amplitude errors onto the signal passing through it. The result is AM/AM (amplitude dependent gain) and AM/PM (amplitude dependent phase) distortion, which generally gets worse, when approached near compression region [4]. It is highly dependent on the amplitude of signal and is directly related to the amplitude

component of modulation. The effectiveness of any solution is dependent on the amplifiers characteristics, which should be determined in the first instance. If the amplifier exhibits gentle compression characteristics then there is a good chance of improving its efficiency by linearisation [5]. Therefore, this synopsis focuses on employing a lineariser in order to enhance the behavior of an amplifier.

1.2 Amplitude distortion

Distortion can be thought of as the creation of undesired signals at frequencies not contained in the original signal. Amplifier linearity can be considered a measure of how closely the input-output transfer response of an amplifier resembles a straight-line. When an amplifier's input level increases by a certain percent, its output should increase by the same percent. An ideal amplifier (Figure 1.2) would have transfer characteristics, where the output voltage would be a scalar multiple of the input voltage, that is

$$V_{out}(t) = W_1 V_{in}(t) \tag{2}$$

Where W_1 is the gain of amplifier.

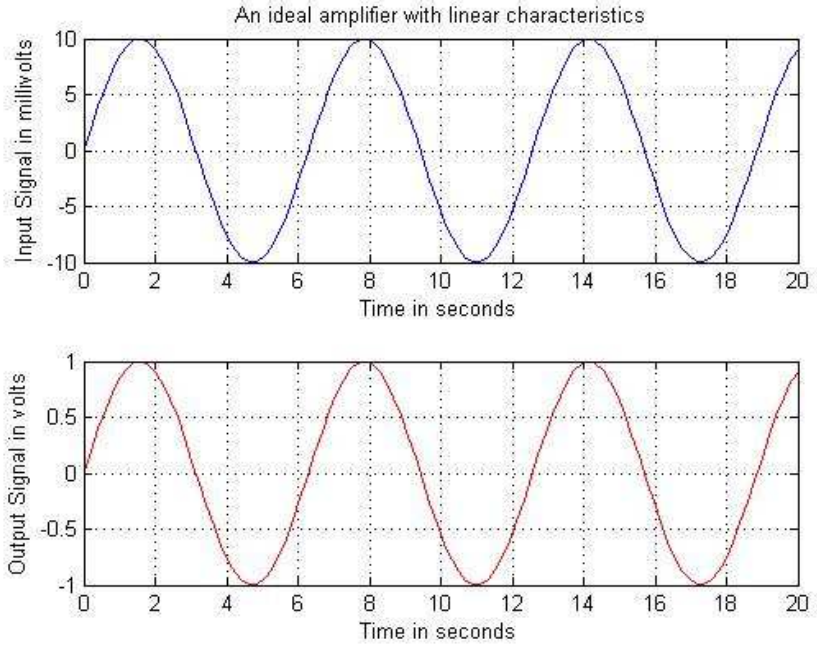


Figure 1.2: Response of an ideal amplifier with sine wave as an input signal.

The output wave from such an amplifier will be identical to that of the input signal and no new (additional) frequency components will be introduced either within or outside of the amplifier bandwidth. It can be seen from Figure 1.2, where the gain W_1 of the amplifier is taken as 100, the input signal is in millivolts and the output signal is in volts.

As seen from Figure 1.2, there is no amplitude distortion and phase distortion in an ideal amplifier. However in practice, there is a distortion (both amplitude and phase), which reduces the efficiency of the amplifier. This distortion is due to the non-linear response of an amplifier as shown in Figure 1.3. Due to the non-linear characteristics of the amplifier (here the transfer function is tangent hyperbolic function), there is a amplitude distortion in the output of amplifier which reduces the performance of the amplifier. The effect of this non-linearity can be reduced using a perturbation based approximation method. This is focus of research reported in this synopsis and will be discussed further.

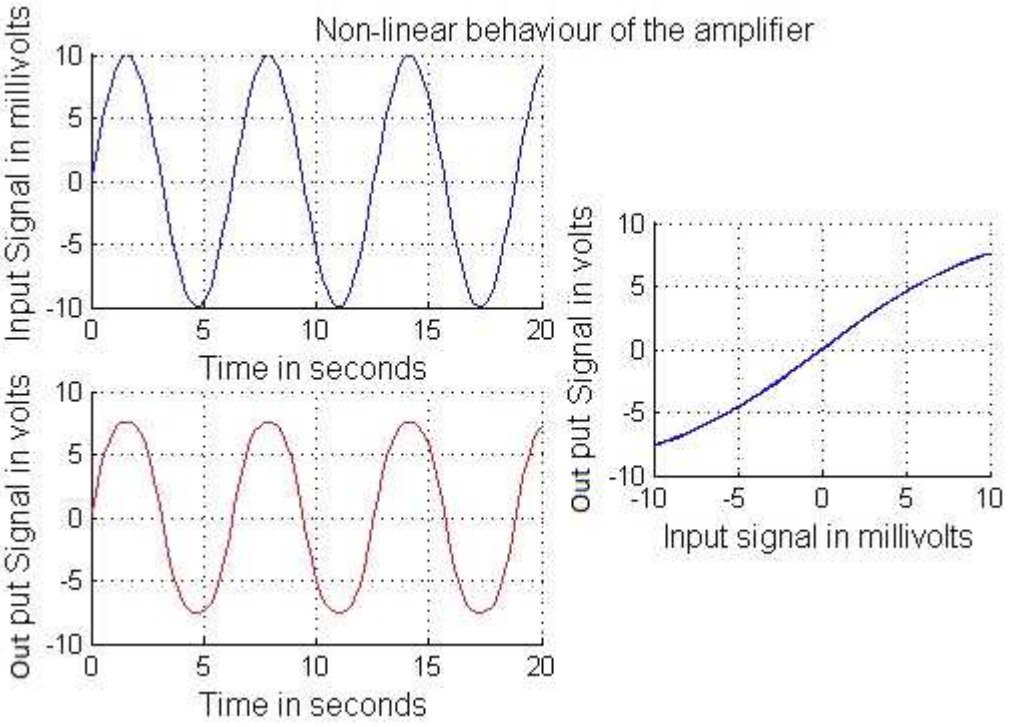

Non-linear behaviour of the amplifier
Input Signal in millivolts
Time in seconds
Out put Signal in volts
Time in seconds
Out put Signal in volts
Input signal in millivolts

Figure 1.3: Response of a non-linear amplifier with sine wave as an input signal.

1.3 Effect of non-linearity

To investigate the effect of non-linearity, consider the non-linearity (i.e. deviation from the straight line) of the amplifier represented by power series as:

$$V_{out}= W_0+W_1V_{in}+ W_2V_{in}^2 + W_3V_{in}^3....+ W_nV_{in}^n \quad (3)$$

Where V_{in} is the input signal, W_1 is the gain of the amplifier and V_{out} is the output of the amplifier.

Assuming the fifth order (n=5) polynomial (for simplicity):

$$V_{out}= W_0+W_1V_{in}+ W_2V_{in}^2 + W_3V_{in}^3+ W_4V_{in}^4+ W_5V_{in}^5 \quad (4)$$

Where, W_0, W_1, W_2, W_3, W_4 and W_5 are the coefficients of polynomial. The higher order terms causes the non-linearity in the amplifier characteristics. The effects of these terms can be reduced by changing the value of their coefficients in such a way that overall sum cancels each other leaving only the first order term, i.e., the linear amplifier. This can be done using the perturbation based approximation method. For this purpose, the first step is to study the effect of each term of the polynomial on the signal distortion. In the beginning it is assumed that there is only amplitude distortion due to the non-linearity of amplifier.

1.4 Effect of second order polynomial

Consider the simplest form of the amplitude nonlinearity by taking the second order polynomial and assuming $W_0=0$, $W_1=10$ and $W_2=2$. So the transfer characteristics of amplifier become:

$$V_{out}= 10V_{in}+ 2V_{in}^2 \quad (5)$$

When a single carrier input signal represented by sine wave is substituted into the above expression the output waveform will contain original sine wave and harmonic distortion products. The harmonics can be eliminated by filtering and do not pose a problem except for the wide band communication applications. However, when more than one carrier is

present, beat products are produced in the vicinity of input signals, which are known inter-modulation distortion products. The output of the above non-linear characteristics (Eqn. 5) of the amplifier is shown in Figure 1.4 below.

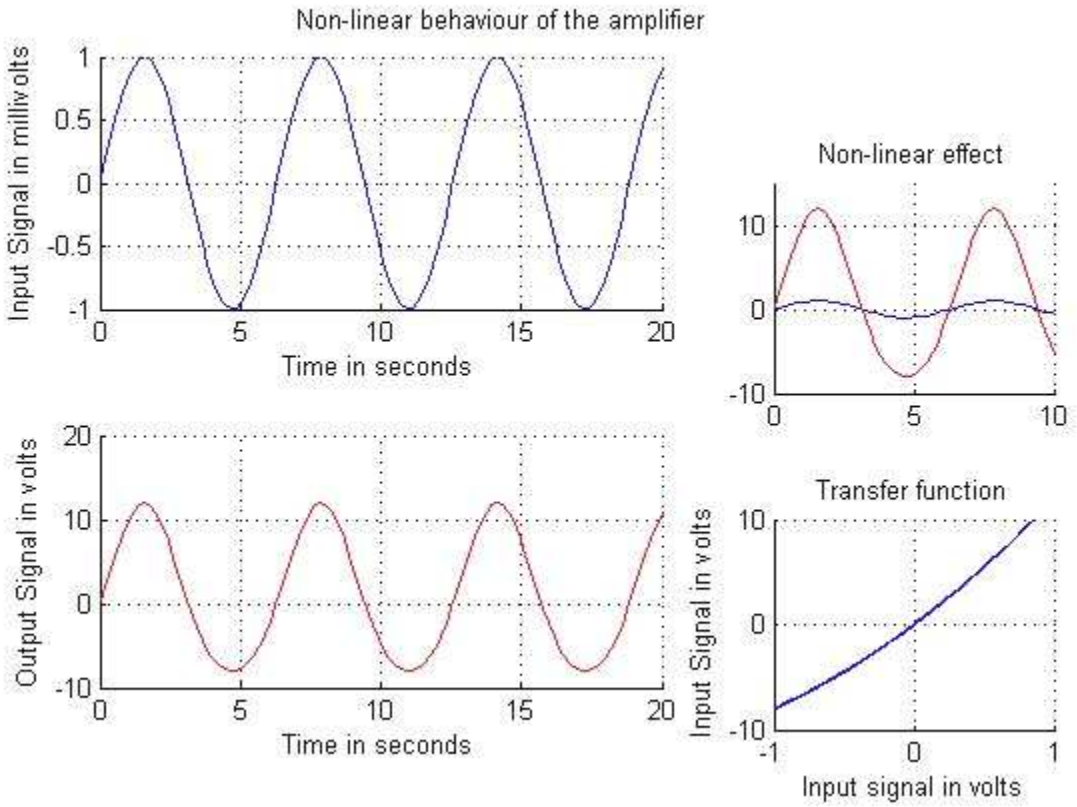


Figure 1.4: Response of a second order non-linear amplifier with sine wave as an input signal.

As seen from the results in Figure 1.4, there is an amplitude distortion. The transfer characteristics of the amplifier are non-linear and are bending upwards towards the y-axis and continue towards infinity. Because of this, in the frequency domain, a second signal component will appear at twice the original frequency and thus gives rise to the term second harmonic distortion. The amplitude of the second harmonic component will increase in proportion to the square of input signal. But the amplitude of the input fundamental frequency component will increase in proportion to the fundamental gain W_1. As a result

the amplitude of the second harmonic will increase at a greater rate than the fundamental component.

1.5 Effect of third order polynomial

Consider the transfer characteristics of the amplifier represented by the third order polynomial:

$$V_{out} = W_1 V_{in} + W_3 V_{in}^3 \qquad (6)$$

It is to be noted that in the beginning, the response of each term and its co-efficient is studied, so the second order term is missing here.

Taking W_1=10 and W_2=-3, the results for this polynomial is shown in Figure 1.5 below.

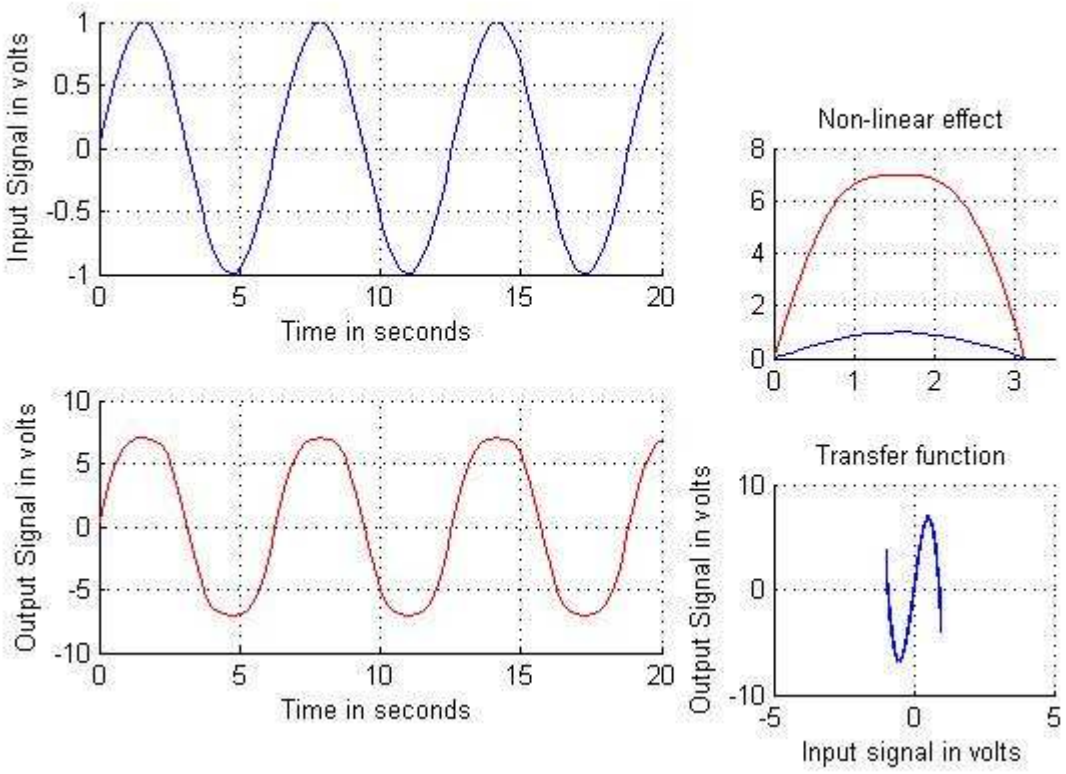


Figure 1.5: Response of a third order non-linear amplifier with sine wave as an input signal.

It is observed from the third order transfer characteristics shown in Figure 1.5 that the output wave shape is now symmetrical above and below the horizontal axis and that a term at three times the original input signal frequency spectrum will appear in the spectrum.

This signal is the third harmonic and gives rise to its description as third harmonic distortion. Further note that no DC components exist for third order distortion unlike that present with second order polynomial. Also, it is important to note that the coefficients of third order term are negative. If the coefficients are positive, then the resulting transfer characteristics will not resemble to that of the amplifier and points towards infinity on both sides of the axis. The height of the point above the axis, where the transfer characteristics starts bending in the opposite side depends upon the magnitude of the third order term.

From Figure 1.6, it is also observed that at low levels, the magnitude of the fundamental component increases almost linearly with the input signal level; however, it then begins to deviate from linear characteristics and eventually decreases again.

1.6 Effect of fourth order polynomial

Consider the following fourth order polynomial representing the transfer characteristics of the amplifier:

$$V_{out} = W_1 V_{in} + W_4 V_{in}^4 \qquad (7)$$

Taking $W_1=10$ and $W_4=2$, the response of the amplifier with these characteristics is shown in Figure 1.6 below.

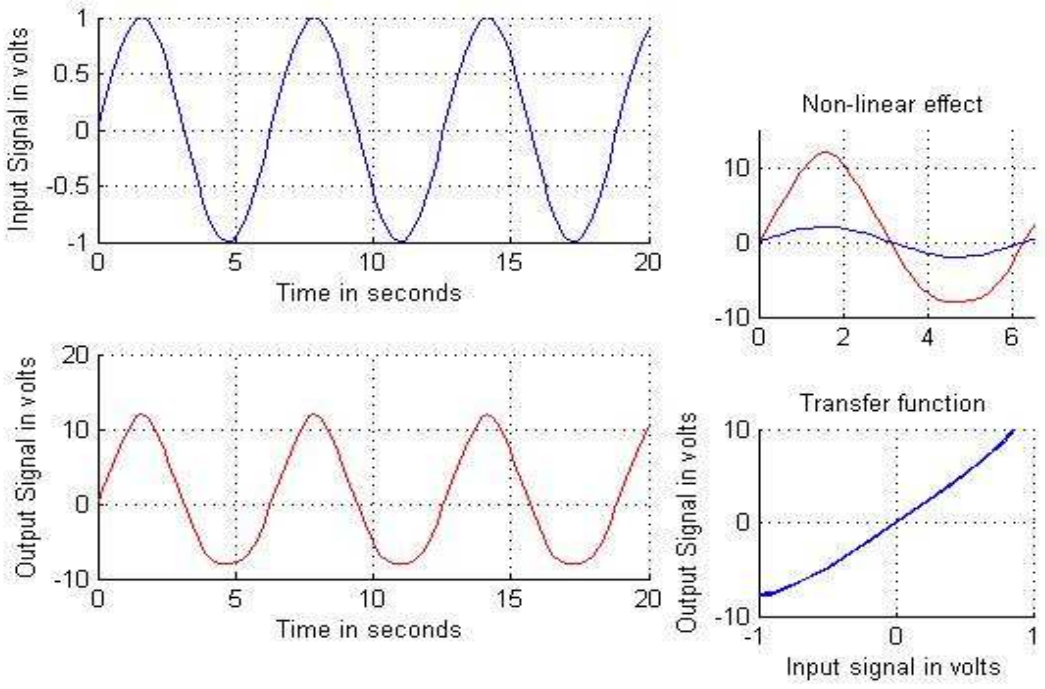


Figure 1.6: Response of a fourth order non-linear amplifier with sine wave as an input signal.

It is observed that the results are similar to that obtained with the second order polynomial. Here also the output is not symmetrical about the x-axis (time axis) and there is an amplitude distortion. However, if this transfer characteristics of the amplifier is compared with the second order polynomial characteristics, then the curve (here) is more bend at the edges and is almost linear in between, which causes the lower region of the output wave to be more flat as shown in Figure 1.6. The transfer characteristics are not symmetrical about the x-axis, which gives the DC shifting in the output. Thus, the fourth order polynomial also gives similar results as second order polynomial, only difference being that here the distortion is more and the output wave-shape is more asymmetrical.

1.7 Effect of fifth order polynomial

Consider the transfer characteristic of amplifier represented by the fifth order polynomial:

$$V_{out}= W_1V_{in}+ W_5V_{in}^5 \quad (8)$$

Taking W_1=10 and W_5= -1, the results for the fifth order term is shown in Figure 1.7. From Figure 1.7, the results obtained are similar to that in Figure 1.6. The difference being that there is more distortion and also it can be observed from the transfer characteristics, that the curve is more sharply bend which increases distortion in the output signal.

Combining all the terms to have:

$$V_{out}= W_1V_{in}+W_2V_{in}^2+W_3V_{in}^3+W_4V_{in}^4+W_5V_{in}^5 \quad (9)$$

Taking W_1=10, W_2=2, W_3=-3, W_4=2 and W_5=-1, the above equation becomes

$$V_{out}= 10*V_{in}+2*V_{in}^2-3*V_{in}^3+2*V_{in}^4-V_{in}^5 \quad (10)$$

The results obtained are shown in Figure 1.8. It is observed that output wave-shape is not symmetrical. There is DC shifting and the distortion is high. The transfer characteristics are not symmetrical about the *x*-axis so the lower portion of the output is more flat. It can also be observed that there is a bend in the lower portion of the output wave. It may be explained on the fact that the transfer characteristics are also not symmetrical about the *y*-axis, as there is slight shifting of the curve upwards which causes the lower portion of the curve to be bend in between.

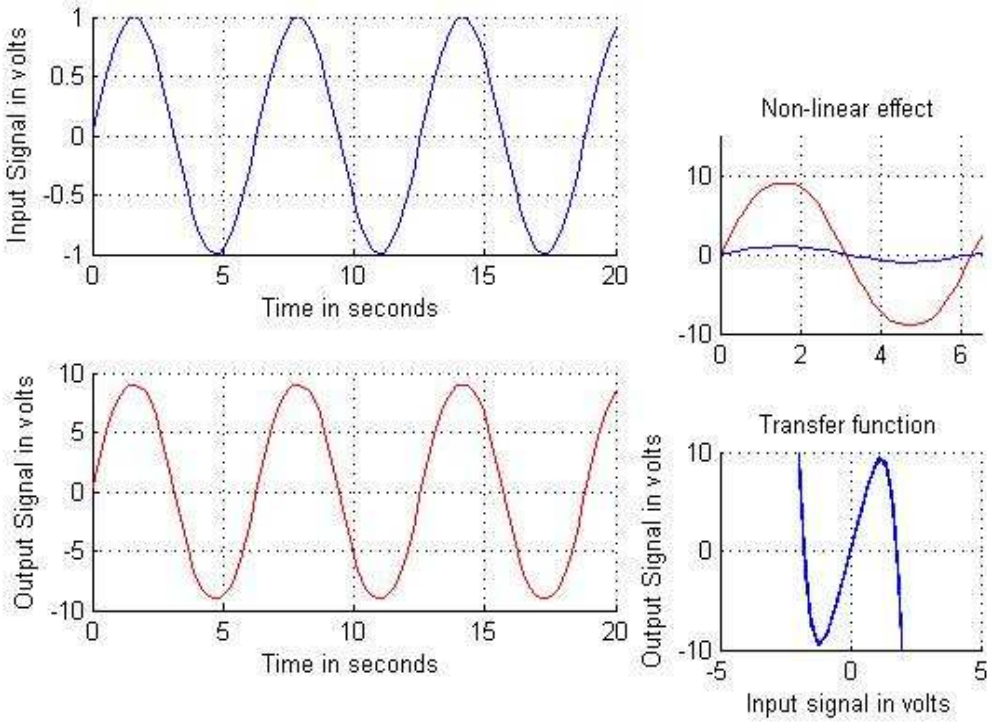


Figure 1.7: Response of a fifth order non-linear amplifier with sine wave as an input signal.

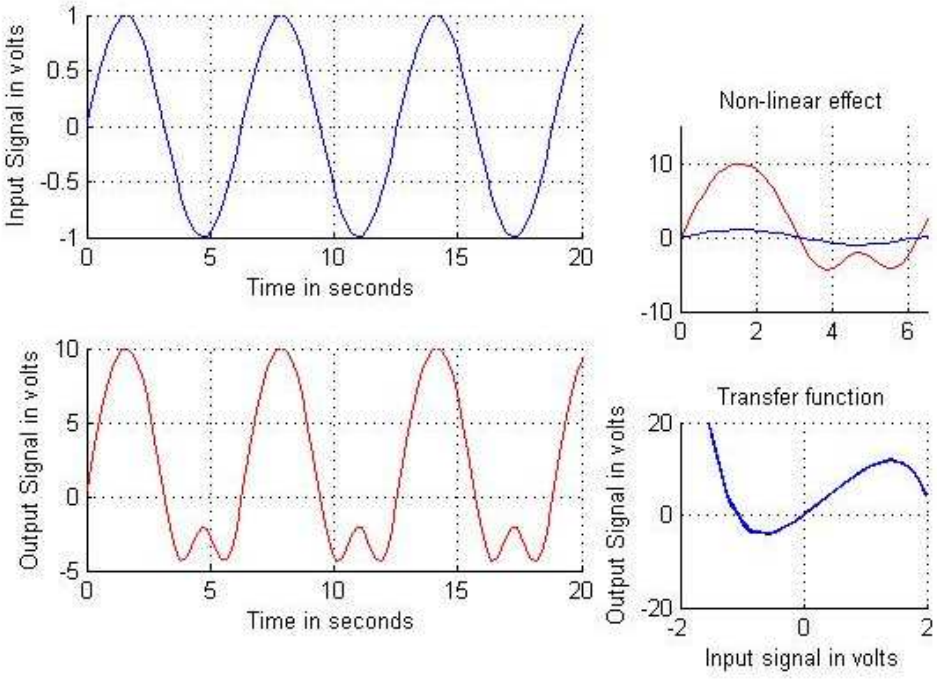


Figure 1.8: Response of an amplifier with the fifth order non-linear characteristics taking all the terms and sine wave as an input signal.

From the above results, it can be interpreted that the characteristics of the even order terms does not reach saturation, i.e., the fourth order (even order) term is increasing upwards, but the odd order terms reaches saturation point both in the case of the third order and fifth order polynomial and then bend downwards. It can also be observed from the characteristics of amplifiers that the characteristics match the tangent hyperbolic curve and reaches saturation. So, it can be concluded that the polynomials, representing the characteristics of amplifier does not have the even order terms (as these does not reach saturation) except the zero order term which represents DC shifting in the output curve. The polynomial approximation up to fifth order for the tangent hyperbolic curve is given by:

$$\tanh(t) = 0.0725*t^5 - 0.3076*t^3 + 0.9971*t \quad (11)$$

Also, in this case the coefficients of each of the alternate terms after the third order terms change the sign alternatively. Therefore, the characteristics of the amplifier can be represented as:

$$Y(x) = w_0 + w_1 x + w_3 x^3 + w_5 x^5 + \ldots + w_{(2*n-1)} x^{(2*n-1)} = w_0 + \sum_{i=0}^{n} w_{(2*i-1)} x^{(2*i-1)} \quad (12)$$

1.8 Key conclusions

The key conclusions drawn from the above discussion are:

1. When more than one carrier is present at the input of a high power amplifier, it leads to production of IMDs. Ideally, these IMDs must be reduced to a level, where all the connecting systems can operate satisfactorily. It is because these IMD products cause significant level of interference in high power broadcast transmitters.
2. HPA, depending upon its characteristics, can impart both phase and amplitude errors onto the signal passing through it.
3. Lineariser can be employed to enhance the behavior of an amplifier if an amplifier exhibits gentle compression characteristics. Due to non-linear response of amplifier, amplitude distortion in the output reduces its performance. The effect of non-linearity can be reduced using perturbation based approximation method.
4. The coefficients of higher order terms in the input-output relation can be changed in such a way that the overall sum cancels each other leaving only the first order term i.e. linear amplifier.

5. Due to second order polynomial, a second component appears at twice the original frequency and thus gives rise to second harmonic distortion.
6. Due to third order polynomial, a component at three times the original input signal frequency appears and thus gives rise to third harmonic distortion.
7. Fourth order polynomial gives similar results as second order with a difference that the distortion is more and the output wave shape is asymmetrical.

From the above discussion and key conclusions, it can be inferred that linearisers are very important for reducing the signal distortions produced due to the non-linear behavior of the amplifier. Several researchers and engineers have worked in this area. However, none of these is able to fully solve this problem. This makes the author motivated to perform research in this area.

1.9 Problem formulation

A solution to the above problem of signal distortion due to the non-linear behavior of the high power amplifiers is by using the adaptive linearisers. In practice, lineariser is usually an extra component configured into a subassembly or box, added to the design of a conventional amplifier. Several engineers and researchers have proposed and designed many models of linearisers such as feed-forward, feed backward, envelope feedback system, cartesian feedback system and predistortion. However, these are associated with several drawbacks. The most common of these include: phase delay, reduction of the achievable bandwidth, degradation of gain and phase margin, low power efficiency for low input signals. Among the above mentioned linearisation techniques, pre-distortion is the most commonly used. It is based on the concept, to insert a nonlinear module between the input signal and power amplifier. The nonlinear module generates IMD products that are in anti-phase with the IMD products produced by the power amplifier. These anti-phase IMD signals will ideally cancel the IMDs produced by HPA; and thus, resulting in a better overall linearised behavior of the whole system. However for highly linear applications, the adjustment in settings of lineariser is usually very critical. As a result of the parameter sensitivity of linearisers, much effort has been devoted to the development of linearisers that can automatically adapt to environmental and stimulus changes. These types of

linearisers are called as adaptive linearisers. In research reported in this synopsis, a novel idea is presented to design an adaptive lineariser using perturbation based method; whereby, the characteristics of lineariser are represented by a non-linear polynomial. This approach is expected to give a better overall linear amplification system. This is discussed further in this synopsis and forms the principle research reported in this synposis.

1.10 Research aim

To design adaptive lineariser for RF wide band amplifers and to assess its performance in terms of minimising the non-linearity of HPA and other parameters.

1.11 Objectives

The key objectives to overcome the signal distortion problem (as reported above) are summarized below:

1. Review of existing (and in developing stage) various linerisation techniques used with high power amplifiers.
2. To investigate the associated problems of above reviewed techniques and to present the design of an adaptive predistortion based lineariser.
3. To develop a methodology for implementing the above mentioned predistortion based adaptive lineariser and to adjust its parameters.
4. To design and implement the perturbation based algorithm, in order to design a predistortion based adaptive lineariser, and to adjust the coefficients of polynomial representing the characteristics of lineariser.
5. To perform a pilot study trial of the developed model, aiming to investigate its efficiency in reducing the non-linearity of high power amplifiers.

CHAPTER 2

A REVIEW OF LINEARISATION TECHNIQUES

2.1 Linearisation techniques

Linearisation is a systematic procedure for reducing an amplifier's distortion. There are different ways of linearising an amplifier. The key technologies are:

1. Feed-forward linearisation
2. Feed-backward linearisation
3. Pre-distortion linearisation
4. Adaptive linearisation

Making a choice of which linearisation technique is to employ for a particular application involves tradeoffs of complexity, degree of IMD suppression and bandwidth. Usually extra components are added to the design of the conventional amplifier, which are configured into a subassembly or a box called as a lineariser. The linearisation techniques can be categorized into three groups [6,7,8].

The first group includes methods where a signal having a varying envelope is fed to the power amplifier with the non-linear characteristics, so that IMD (intermodulation distortion) is generated. The IMD can be cancelled either by isolating the IMD component and subtract it from the amplifier output (Feed-forward) or by shaping the input signal in such a way that the IMD is cancelled at the output of the amplifier (pre-distortion or the negative feedback).

The second group contains the methods in which a signal with a constant envelope is fed into the amplifier having the non-linear characteristics. One method is based on splitting the input signal having varying amplitude and phase into two signals having a constant envelope and a varying phase [7, 8]. These signals are amplified separately without distortion and recombined to form an amplified replica of the input signal. In yet another scheme the input signal is split up into its polar components, i.e. amplitude and phase. Only the phase information is carried by the input signal to the amplifier. The amplitude is

applied by modulating the voltage supply of the amplifier (envelope elimination and restoration).

The third scheme is based on the difference (error) between the output of the ideal amplifier and the practical amplifier, both having the same gain. The gain or the transfer characteristics of the amplifier is expressed into the polynomial form and depending on the value of the error, the coefficients of the polynomial is changed so that the distortion or the error can be minimized. An overview of these techniques is given below.

2.2 Feed forward linearisation

In 1920's, H.S. Black of Bell telephone laboratories invented two schemes for reducing amplifier distortion, namely feed-forward and negative feedback [4,5,6]. His idea for feed forward was simple; reduce the amplifier output to the same level as input and subtract one from the other to leave only the distortion generated by the amplifier. Amplify the distortion with a separate amplifier and then separate it from the original amplifier to leave only a linearly amplified version of the input signal. But in the negative feedback system, an attenuated version of the amplifier output signal is fed back to the input in anti-phase and is combined with the input signal. The same amplifier is used to amplify the difference signal and thus providing a linearly amplified version of the input signal. The feed forward technique required perfect gain match in the different signal paths and Black reported that the gain of the amplifier have to be constantly readjusted. Another reason is that the feed forward system is complex in comparison to the negative feedback system. The basic principle of the feed forward system is described in Figure 2.1 below.

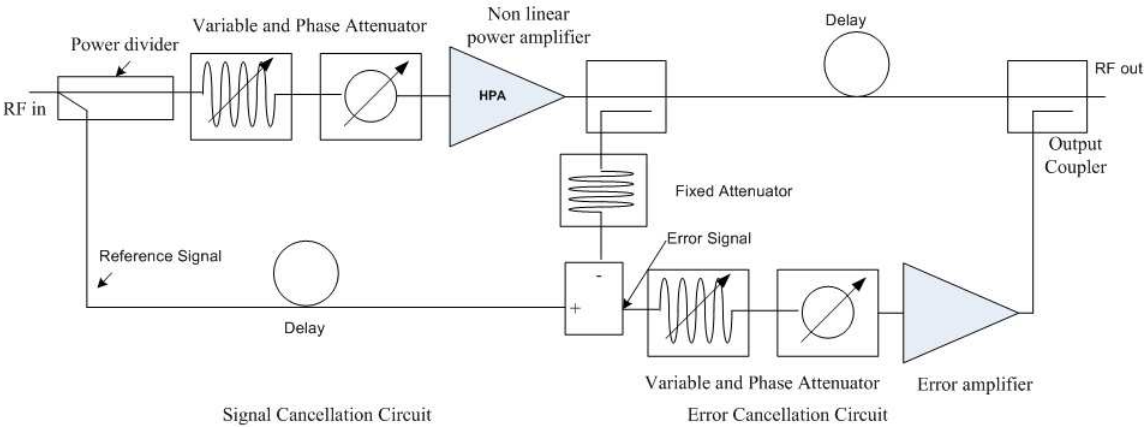


Figure 2.1: Basic feed forward circuit.

It consists of two circuits, the signal cancellation circuit and the error cancellation circuit. The purpose of the signal cancellation circuit is to suppress the reference signal from the main amplifier (PA) output leaving only amplifier distortion, both linear and non-linear, in the error signal. In order to suppress the reference signal, the values of the sampling coupler and fixed attenuation are chosen to match the gain of the main amplifier so that the PA output signal is reduced to approximately the same level as reference signal. The variable phase shifter ahead of the PA is then adjusted to place PA output in anti-phase with the reference.

The variable attenuation serves the fine tuning function of precisely matching the level of the PA output and the reference. The delay line in the reference branch, necessary for the wideband operation, compensates for the group delay of the main amplifier by time aligning the PA output and reference signal before combining [8].

In order to suppress the error signal, the gain of the error amplifier is chosen to match the sum of the values of the sampling coupler, fixed attenuator and output coupler so that the error is increased to approximately the same level as the distortion component of the PA output signal. The variable phase shifter ahead of the error amplifier is then adjusted to place the error in anti-phase with the PA output. The variable attenuation, again serves the fine tuning function of the precisely matching level of the error signal and the distortion component of the PA output. The purpose of the delay line is same as in case of the error cancellation loop. The error amplifier must be chosen that it linearly amplifies the error signal while still providing the required output power. This usually dictates the use of a class A amplifier with sufficient back off.

For the proper operation of the feed forward circuit the proper adjustment of the attenuation and phase in the signal and error cancellation circuits has to be done so that a good IMD suppression is maintained over time.

Variations of the component characteristics with temperature and time as changes in operating conditions such as input power level and supply all necessitate readjustment. Fixed or manually controlled amplitude and phase matching networks are usually not sufficient to preserve reasonable distortion suppression. Components ageing, temperature drift, change of operating frequency etc cause variations that require automation control of

the amplitude and phase matching networks. For these reasons Black himself essentially abandoned feed forward in favor of feed back.

2.3 Feedback linearisation

A basic block diagram of the negative feedback system is shown in Figure 2.2 below. The distortion generated by the amplifier is reduced by an amount equal to the return difference of the feedback system F=1+βA [9]. The gain is also reduced by the same amount.

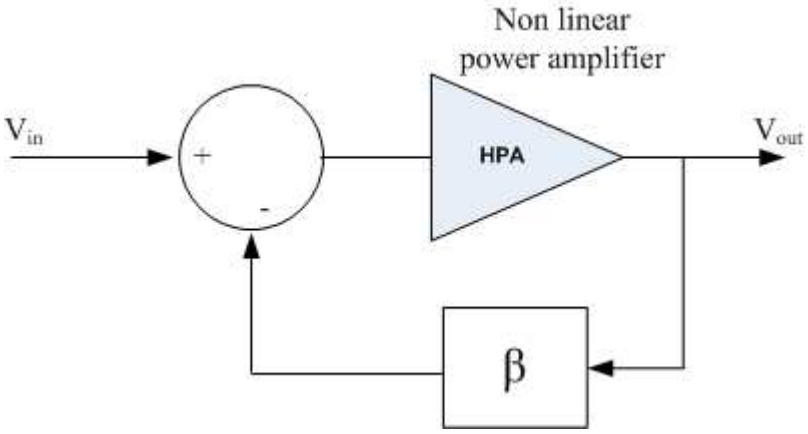


Figure 2.2: Basic negative feedback circuit.

Now the overall gain, G, is given by

$$G = V_{out} / V_{in} = A/(1 + A\beta) \qquad (13)$$

So for the large values of Aβ, the gain approximates to $G = 1/\beta$. So, the gain of the amplifier must be sufficiently high to allow for a high gain and high distortion suppression. At high frequencies, the gain becomes rather moderate so this is not a solution for the high frequency applications. The gain can be increased by cascading the number of these devices but as the number of stages increases the system stability will decrease. Oscillation is also another major problem. Another problem is that the calculated benefits of the negative feedback only applies at 180 degrees point, and erodes away either side, as the phase response rotates away from the ideal value (180 degrees point). The block diagram of the envelope feedback system is shown in Figure 2.3. In this approach an amplifier input and the output signal is detected and low pass filtered and the resulting base band signals are compared.

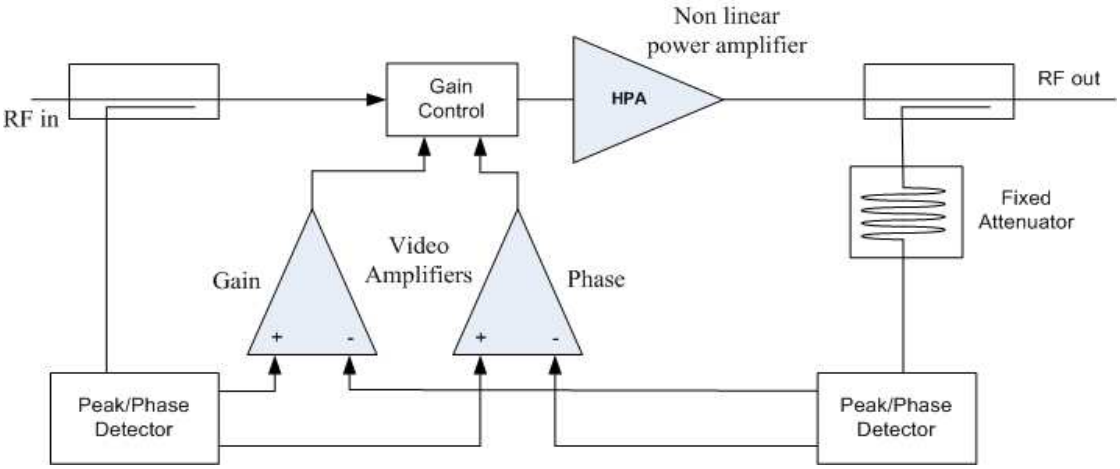


Figure 2.3: Basic envelope feedback circuit [9].

The input and the output detectors are assumed to be perfectly matched and the associated coupling and attenuation values are chosen such that the linear gain of the PA is cancelled, so that detectors receive the identical RF signal levels in linear operation [9]. The detectors are fed into a video differential amplifier, generating an error signal which drives an input attenuator in the appropriate sense to restore the error sensed by the detectors. Superior linearity can be obtained by correcting both the amplitude and phase.

The fundamental limitation of the envelope feedback system is just the same as the direct feedback system i.e. phase delay around the control loop. Its principle limitation is its inability to handle wideband signals. In practice, it is difficult to make a feedback system respond to signal envelope changes much greater than several MHz, because of the delay (Δt_s) of the amplifier and associated signal processing components. The signal bandwidth must satisfy the following equation for significant correction:

$$BW < 1/(4\Delta t_s) \qquad (14)$$

Thus, the total delay must be less than 25ns for a 100MHz bandwidth.

2.4 Cartesian feedback

Cartesian feedback was introduced by Cox. The block diagram of the Cartesian feedback circuit is shown below in Figure 2.4 [4,5]. The output of the amplifier is synchronously demodulated and compared with the source signal to obtain the error signal. The error

signal is fed into the loop filter followed by the up conversion in a quadrature modulator before it finally reaches the power amplifier.

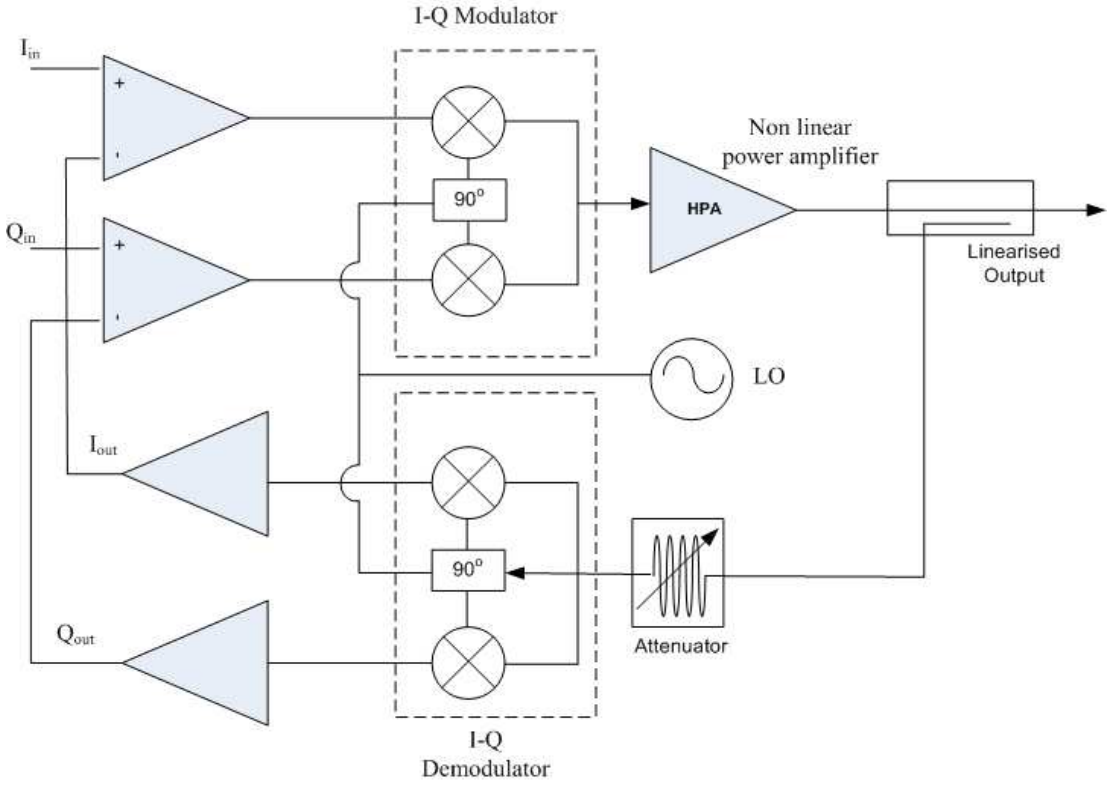


Figure 2.4: Basic Cartesian loop linearisation system [4].

Cartesian feedback is often used with the quadrature phase shift-keyed (QPSK) modulation. The problem in the Cartesian feedback is the delay in the mixers and the reduction of the achievable bandwidth. Noise and linearity of the feedback mixers affect directly the output. Another problem being that the characteristics of the amplifier effectively degrades the gain and the phase margins of the loop. Another problem is the phase shift that occurs in loop which causes the need of an automatic phase adjuster so as to maintain stability, as can be seen in Figure 2.4. Another disadvantage is that the power efficiency is low for the low input signals, when the amplifier is fed with the signal having a varying envelope. An advantage of the Cartesian feedback is that the BWs of the in phase and quadrature components are approximately equal.

2.5 Polar feedback system

The technique is illustrated in Figure 2.5. The input signal is an intermediate frequency signal. This signal is split up in its polar components, amplitude and phase and compared with their counterparts of the amplifier's output signal.

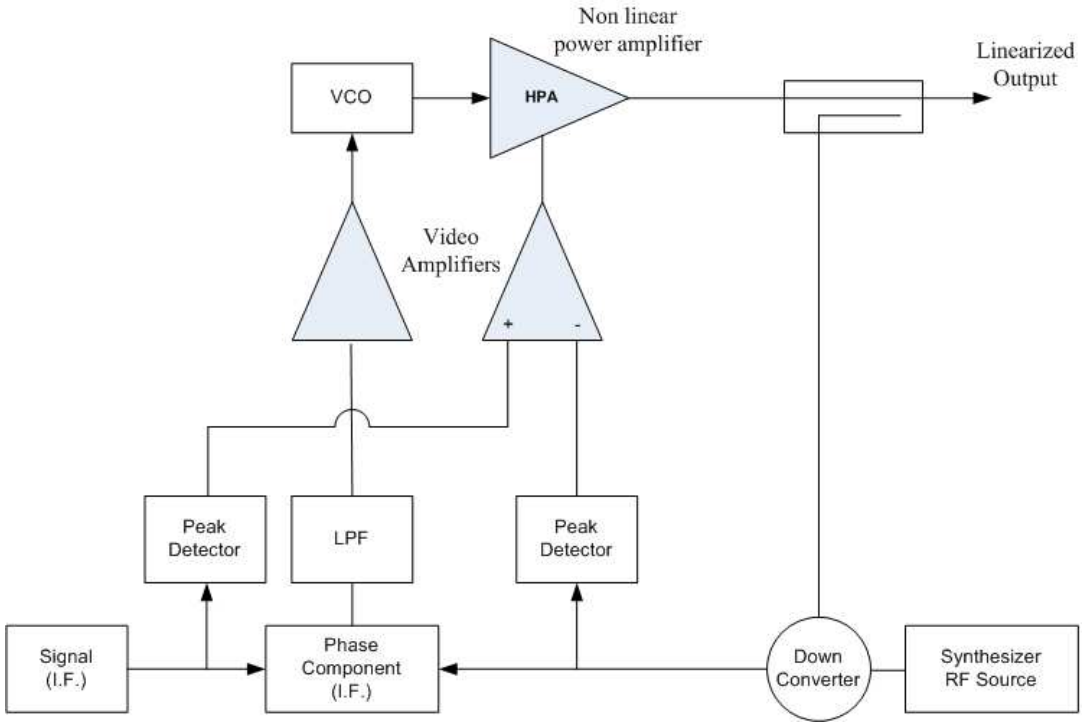


Figure 2.5: Basic polar loop system.

The resulting phase error signal controls a VCO that feeds the amplifier with a constant envelope but phase modulated signal. Thus, a phase locked loop is used to track the phase and a classical feedback circuitry to track the amplitude. The channel frequency is set by the local oscillator in the feedback chain. In the polar feedback system then bandwidth of the phase components is much greater than the bandwidth of the amplitude components.

2.6 Pre-distortion linearisation

Predistortion (PD) linearisers have been used extensively in the microwave and satellite applications because of their relative simplicity and their ability to be added to existing amplifiers as separate stand alone units [10,11,12,13,14]. PD's provide a viable

improvement in linearity near saturation, but are difficult to apply in applications requiring high linearity [15,16]. PD linearisers generate a non-linear transfer characteristics, which is the reverse of the amplifier's transfer characteristics in both magnitude and phase as shown in Figure 2.6 [17].

The basic principle is that if the IMD products produced by the linearisers is made equal in amplitude and 180 degrees out phase with the IMD's generated by the amplifier, the IMD's will cancel. This condition occurs when the gain of the linearised amplifier remain constant with change in power levels.

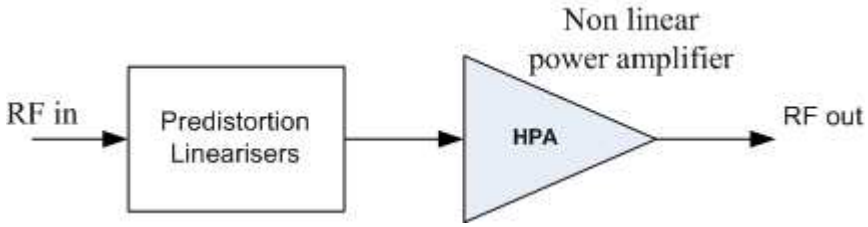
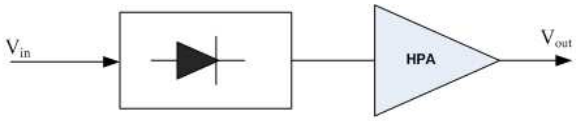
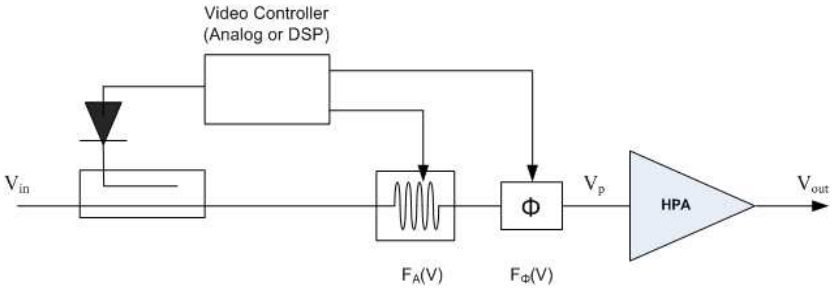


Figure 2.6: Basic predistortion lineariser for high power amplifiers.

There are two basic methods by which the input signal to the power amplifier can be pre-distorted [18,19]. These are shown in the Figure 2.7.



(a)



(b)

Figure 2.7: Predistortion linearisers (a) non-linear element and (b) vector modulator.

In the first method, a physical non-linear device is used. This device, usually consisting of one or several diodes, has to be tailored to have the best possible approximation of the

required PD characteristics. An alternative approach is shown in Figure 2.7b. Here the amplifier has a gain and a phase modulator placed on its input, so that the gain and the phase of the PA can be adjusted in accordance with the previously measured non-linear properties. The modulator, therefore, requires a two-dimensional drive signal to perform its task. There are a number of predistortion techniques, such as analog pre-distortion, mapping pre-distortion, polar pre-distortion and complex gain pre-distortion.

The major disadvantage with the predistortion techniques is their complexity and the longer adaptation times like in mapping pre-distortion due to the size of the two dimensional table used.

2.7 Summary

From the above discussion, it can be interpreted that the above mentioned research only partially met the requirements of eliminating the signal distortion due to the non-linear behavior of an amplifier. The various lineariser techniques discussed above do not fully meet the above requirement of reducing signal distortion. Therefore, a new research approach is proposed in synopsis, which focuses on the implementation of an adaptive pre-distortion based lineariser, in which the co-efficents of polynomial representing the characteristics of lineariser are adjusted using the perturbation based algorithm. It will be discussed in detail in Chapter 3.

CHAPTER 3

RESEARCH METHODOLOGY & DESIGN

3.1 Problem formulation

This chapter presents a solution to the problem of signal distortion due to the non-linear behavior of the high power amplifiers. For this purpose, the process of linearisation is used. Linearisation can be referred as a procedure for reducing the distortion produced in the response of an amplifier. In practice, lineariser is usually an extra component configured into a subassembly or box, added to the design of a conventional amplifier. Several engineers and researchers have proposed and designed many models of linearisers such as feed-forward, feed backward, envelope feedback system, cartesian feedback system and predistortion. However, these are associated with several drawbacks. The most common of these include: phase delay, reduction of the achievable bandwidth, degradation of gain and phase margin, low power efficiency for low input signals. Among the above mentioned linearisation techniques, pre-distortion is the most commonly used. It is based on the concept, to insert a nonlinear module between the input signal and power amplifier. The nonlinear module generates IMD products that are in anti-phase with the IMD products produced by the power amplifier. These anti-phase IMD signals will ideally cancel the IMDs produced by HPA; and thus, resulting in a better overall linearised behavior of the whole system. However for highly linear applications, the adjustment in settings of lineariser is usually very critical. As a result of the parameter sensitivity of linearisers, much effort has been devoted to the development of linearisers that can automatically adapt to environmental and stimulus changes. These types of linearisers are called as adaptive linearisers [22]. In research reported in this synopsis, a novel idea is presented to design an adaptive lineariser using perturbation based method; whereby, the characteristics of lineariser are represented by a non-linear polynomial. This approach is expected to give a better overall linear amplification system. This is discussed further in this synopsis and forms the principle research reported in this synopsis.

3.2 Methodology

In the first instance, a review of existing and in developing stage linearisation techniques is presented in Chapter 2. From this review, the depth of problem is analysed, which motivated the author to perform research in this domain. In the next stage, an adaptive lineariser is designed to solve the problem of signal distortion due to the non-linear behavior of amplifier.

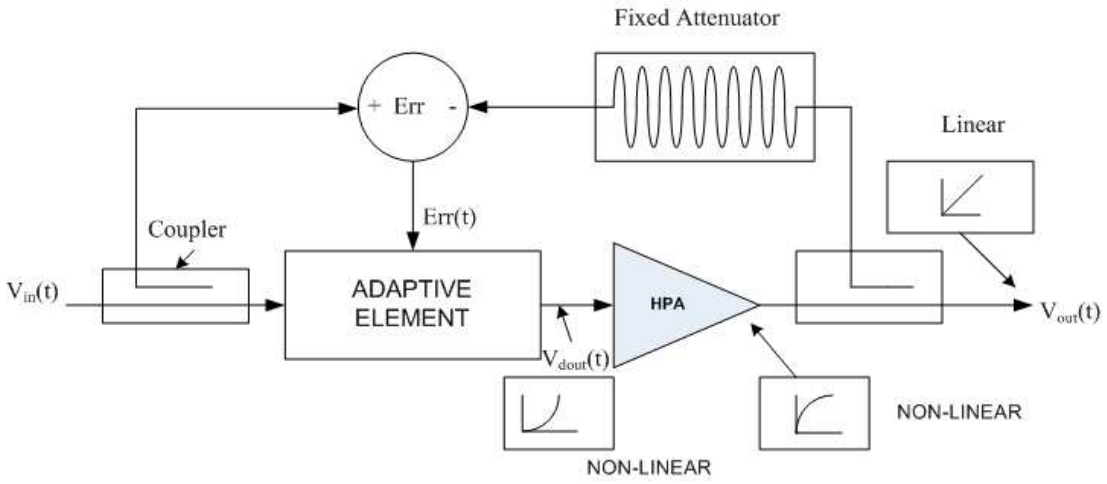


Figure 3.1: Functional block diagram of HPAs using the adaptive linearisation technique.

The principle of an amplifier with an adaptive lineariser is explained in Figures 3.1 and 3.2. Ideally, the transfer characteristics of an amplifier should be linear. However as discussed above, there is non-linearity in the behaviour of HPA, which results in the distortion of signals being amplified. To minimize this distortion, adaptive linearisers are used in this synopsis, aiming to reduce the distortion automatically. It is shown in Figure 3.1, in which the output of non-linear amplifier is fed back to the error detector after attenuating through a fixed attenuator. The error, due to non-linearity of the amplifier, is calculated by taking the difference of the attenuated output of the non-linear amplifier and the input signal. This error is fed to the adaptive lineariser element, which varies the envelope of the input signal V_{in} resulting in the signal V_{dout}. This V_{dout} signal is fed to the amplifier such that Err (t) can be minimised.

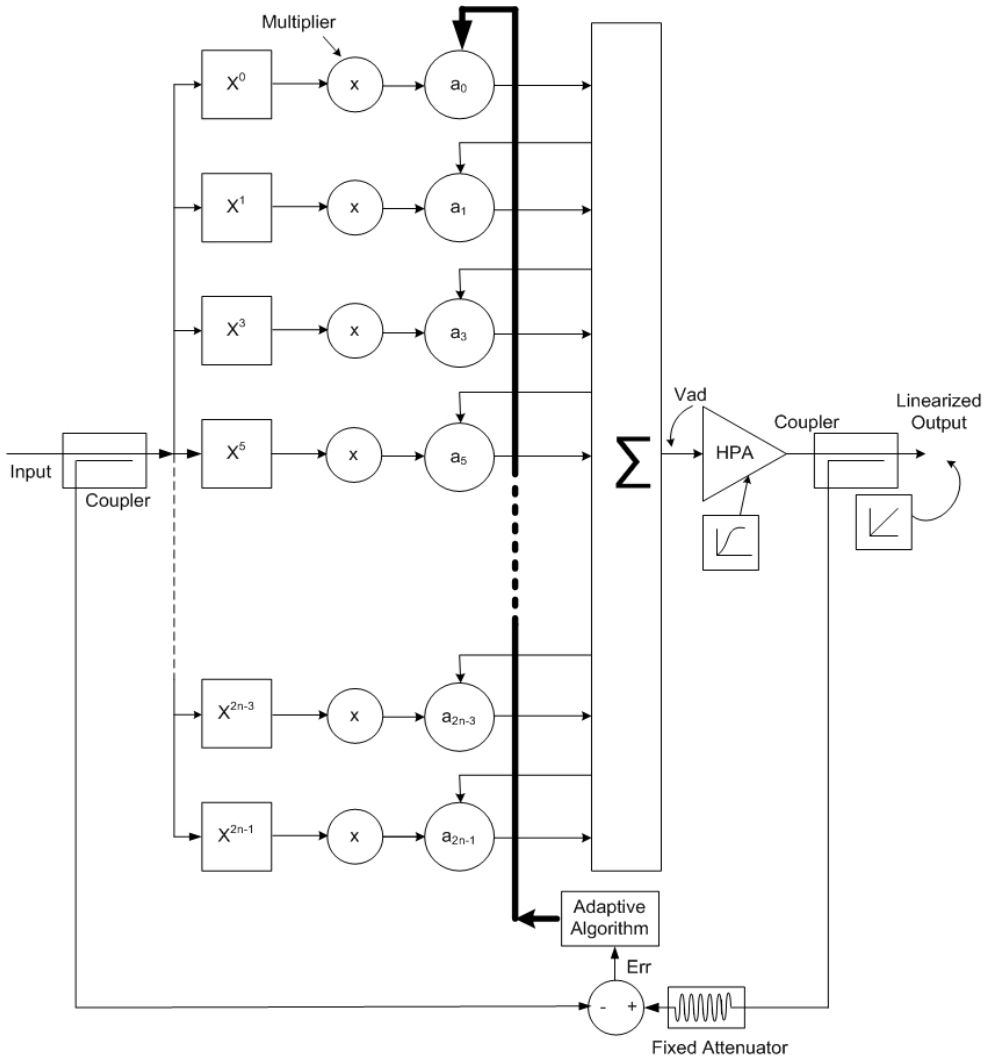


Figure 3.2: Block diagram showing the function of adaptive lineariser.

3.3 Research design

In the proposed adaptive linearisation technique shown above in Figure 3.2, perturbation based approximation method is used. Perturbation theory comprises of mathematical methods that are used to find an approximate solution to a problem which cannot be solved exactly, by starting from the exact solution of a related problem. Perturbation theory leads to an expression for the desired solution in terms of a power series. A power series (in one variable) is an infinite series of the form:

$$f(x) = a_0 + a_1 x + a_2 x^2 + a_3 x^3 + a_4 x^4 + a_5 x^5 + \ldots\ldots\ldots = \sum_{n=0}^{\infty} a_n x^n \qquad (15)$$

Where a_0, a_1, $a_2 \ldots . a_n$ are the co-efficients of the power series and the argument x can be real or complex numbers. If this series is compared by the polynomial representing the transfer characteristics of the lineariser then the second term of the above equation gives the approximation for the gain, while the other terms contributes for the non-linearity of the amplifier. The second term in this power series is the solution of the exactly solvable problem (linear amplifier), while further terms describe the deviation (non-linearity) in the solution, due to deviation from the initial problem (linear problem). The final solution can be given by a series:

$$A_{final} = A_{initial} + A1 + A2 + A3 + \ldots\ldots\ldots\ldots\ldots \qquad (16)$$

Where, A_{final} is the final best approximate solution, $A_{initial}$ would be the known solution to the exactly solvable similar problem, i.e. gain of the linear amplifier and *A1, A2, A3…* represents the "higher orders" which are found iteratively by the systematic procedure described below.

As discussed in Chapter 1, consider the general expression for characteristics of HPA, as represented by:

$$y(x) = w_0 + w_1 x + w_3 x^3 + w_5 x^5 + \ldots + w_{2*N-1} x^{2*N-1} = w_0 + \sum_{i=0}^{N} w_{(2*i-1)} x^{(2*i-1)} \qquad (17)$$

From Figure 3.2, the characteristics of lineriser should be anti-phase to nullify the effect of higher order terms in above equation; representing the non-linear characteristics of an amplifier. Thus, the polynomial representing the characteristics of lineariser can be better expressed as:

$$V_{ad}(t) = a_0 + a_1 V_{in} + a_3 V_{in}^{\ 3} + a_5 V_{in}^{\ 5} + \ldots + a_{2*N-1} V_{in}^{\ 2*N-1} \qquad (18)$$

Thus, the relationship of the transfer function (g) of the amplifier (HPA) and that of adaptive lineariser (f) can be expressed as follows:

$$V_{out}(t) = g(f(V_{in}(t))) \qquad (19)$$

Where function g is the inverse of function f, i.e. if f is the tangent hyperbolic ($\tanh(x)$) function, then g is the inverse tangent hyperbolic ($\tanh^{-1}(x)$) function, so that the output is as linear as possible. Therefore, the coefficients of polynomial representing the

characteristics of an adaptive lineariser are adjusted in such a way that the characteristics of the adaptive lineariser are inverse to that of the non-linear amplifier. If the output signal from such a lineariser is fed to the non linear amplifier then the error approaches to zero and the output of amplifier becomes closely to linear. The coefficients are adjusted using the algorithm based on perturbation based approximation method mentioned below.

3.4 Research design & approach

The major steps involved in this research report are summarized in Figure 3.3 and are detailed below.

- Review of current lineariser technologies and their limitations in meeting the requirements of reducing signal distortions occurring due to the non-linear behavior of HPAs. A pilot stage study has been reported through literature review presented in Chapter 2. This study provided an overview of the research needs.
- Review of current commercial lineariser technologies in use. This study has presented a comparison of current technologies, including their limitations. This comparison has justified the requirement of perturbation based algorithm for producing an adaptive lineariser.
- Develop a perturbation based algorithm for an adaptive lineariser and to evaluate its performance in terms of parameters such as bandwidth, degree of linearity and saturation point. This will help to gain new insights to analyze the feasibility of practically implementing perturbation based algorithms in the domain of linearisers.

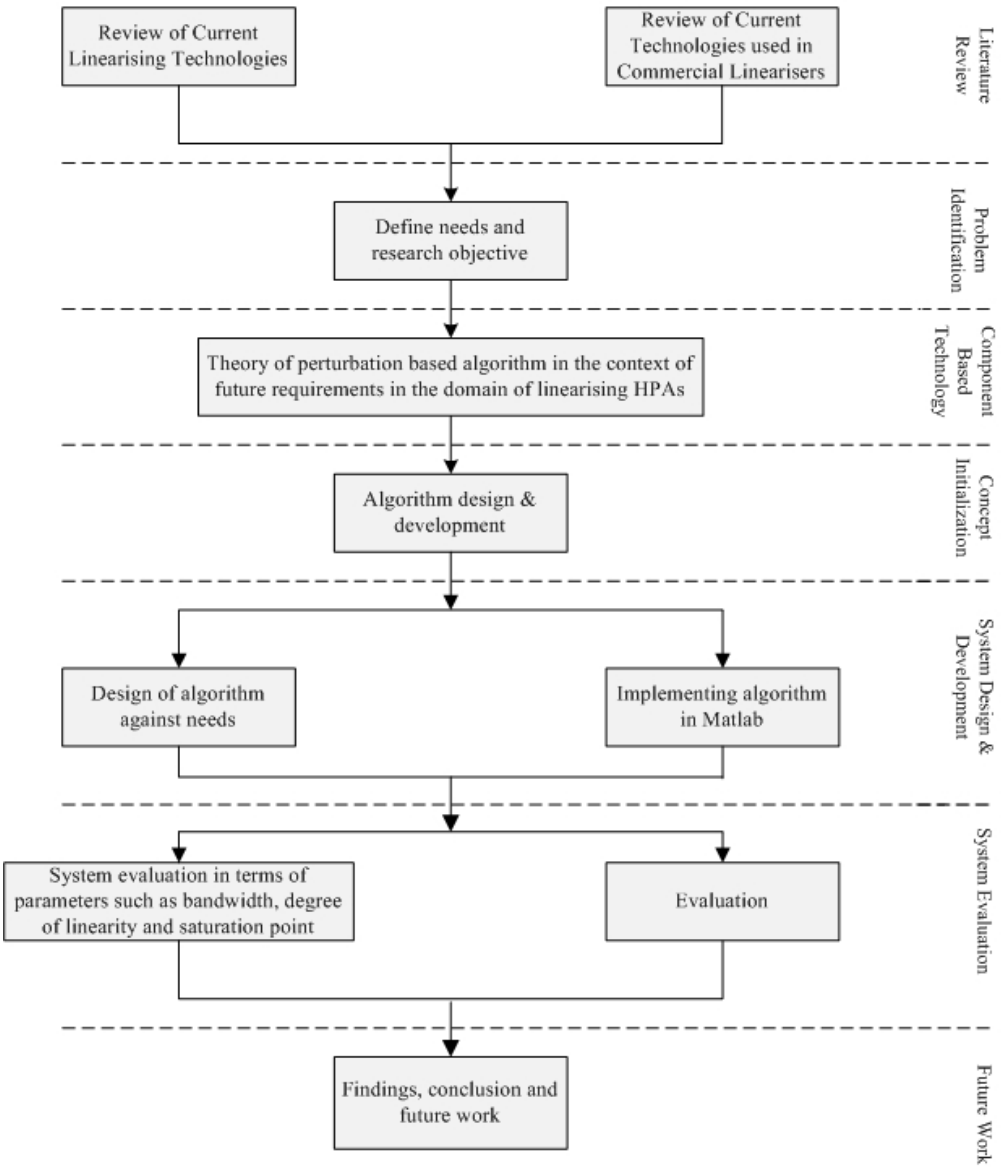

Review of Current Linearising Technologies
Review of Current Technologies used in Commercial Linearisers
Literature Review
Define needs and research objective
Problem Identification
Theory of perturbation based algorithm in the context of future requirements in the domain of linearising HPAs
Component Based Technology
Algorithm design & development
Concept Initialization
Design of algorithm against needs
Implementing algorithm in Matlab
System Design & Development
System evaluation in terms of parameters such as bandwidth, degree of linearity and saturation point
Evaluation
System Evaluation
Findings, conclusion and future work
Future Work

Figure 3.3: Overview of research design approach and methodology.

3.5 Time frame

This chapter presented the design approach to be used in research undertaken through this synopsis. In particularly, it proposed a perturbation based approximation method to design a lineariser which can add predistortions to signal to be amplified prior to feeding to a high power amplifier. The pre-distortions will nullify the effect of non-linear response of the high power amplifier. For this purpose, it proposed an algorithm to design predistortion based adaptive linearisers using the perturbation based approximation method. From the

discussion made in this Chapter and the design methodology presented, it can be inferred that this approach has potential in providing a better solution for the problem of distortion of signal due to the non-linear response of the high power amplifiers. Therefore, the research project focuses on this perturbation based algorithm for linearsizing HPAs. The time frame for completion of this project is mentioned below.

Time Frame (PhD THESIS)	
July 2012-December 2012	Literature Survey, Concept Development and Design
January 2013-June 2013	Implementation and Testing
July 2013-December 2013	Evaluation and Redesign (if any)
	Thesis Writing and Paper presentation
January 2014-June 2014	Thesis Writing

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